PFD:12/19/2000 adaptive equaliz385

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training pilot

WHAT IS CLAIMED IS:

1. A method for adaptive equalization method of a received signal, said method comprising the steps of: Key work

(a) calculating an estimated impulse response value of a channel based on said received signal and a known signal, and storing said estimated impulse-response value in a memory;

(b) calculating the tap coefficient for linear filtering based on said received signal and said known signal;

admitted prior art

(c) linear-filtering said received signal with said tap coefficient; and

(d) calculating a soft decision value from the result of said linear filtering of an information signal of said received signal following at least said known signal and said stored estimated impulse response value.

2. In decoding process for decoding a received code bit b(n) by iterating adaptive equalization of said received code bit b(n) a plurality of times, an adaptive equalization method in which a second soft decision value $\lambda_2[b(n)]$ generated by the soft decision of said code bit b(n) is provided as a priori information, and the equalization of a received signal $\mathbf{R}(n)$ of an M-channel channel through utilization of said a priori information and outputting of a first soft decision value $\lambda_1[b(n)]$ are iterated to adaptively equalize said received signal $\mathbf{R}(n)$, said method comprising the steps of:

in a first round of equalization,

(a) calculating an estimated impulse response value $\mathbf{H}_m(n)$ of said M-channel channel and the tap coefficient $\mathbf{G}(n)$ for linear filtering by the first round of adaptive equalization of said received signal $\mathbf{R}(n)$ of said M-channel, and outputting said first soft decision value $\lambda_1[b(n)]$;

in the second and subsequent rounds of equalization,

(b) calculating the likelihood b′(n) of a code bit sequence b(n) from

said second soft decision value $\lambda_2[b(n)]$ obtained by decoding said code bit b(n) based on said first soft decision value $\lambda_1[b(n)]$;

(c) calculating an estimated impulse response value vector $\mathbf{H}_L(n)$ by approximation approximated with the intersymbol interference component for the code bit b(n) regarded as zero, and calculating the tap coefficient $\mathbf{G}(n)$ for said linear filtering from said estimated impulse response value vector $\mathbf{H}_L(n)$ to update said tap coefficient $\mathbf{G}(n)$;

(d) linear-filtering said likelihood b'(n) of said code bit sequence b(n) with said estimated impulse response value $\mathbf{H}_L(n)$ to generate a replica; a replica of what?

(e) subtracting said replica from said received signal $\mathbf{R}(n)$ to generate a difference signal $\mathbf{R}_c(n)$ without intersymbol interference;

(f) linear-filtering said difference signal $\mathbf{R}_c(n)$ with said tap coefficient $\mathbf{G}(n)$ to generate a signal $\mathbf{Z}(n)$; and

(g) outputting, as the results of said second and subsequent rounds of adaptive equalization, said a first soft decision value $\lambda_1[b(n)]$ updated with said signal $\mathbf{Z}(n)$ and said estimated impulse response value vector $\mathbf{H}_L(n)$.

3. The method of claim 2, wherein said step (a) comprises the steps of:

(a-1) calculating the estimated impulse response value $\mathbf{H}_m(n)$ of said M-channel channel based on said received signal $\mathbf{R}(n)$ and a known signal;

(a-2) calculating said tap coefficient $\mathbf{G}(n)$ by an adaptive algorithm based on said received signal $\mathbf{R}(n)$ and said known signal;

(a-3) linear-filtering said received signal $\mathbf{R}(n)$ with said tap coefficient $\mathbf{G}(n)$; and

(a-4) calculating said first soft decision value $\lambda_1[b(n)]$ from said signal $\mathbf{Z}(n)$ and said estimated impulse response value $\mathbf{H}_m(n)$.

4. The method of claim 2 or 3, wherein said step (c) includes a step of calculating the tap coefficient $\mathbf{G}(n)$ for said linear filtering from said

estimated impulse response value vector $\mathbf{H}_L(n)$ through utilization of a Matrix Inversion Lemma..

5. The method of claim 2, wherein said step (a) comprises the steps of:

(a-1) calculating the estimated impulse response value $\mathbf{H}_m(n)$ of said M-channel channel based on a sample value sequence of said received signal $\mathbf{R}(n)$ and a known signal;

(a-2) determining from said estimated impulse response value $\mathbf{H}_m(n)$ whether the received signal power is larger than a predetermined reference value, and, if larger than said reference value, storing the corresponding path as an effective path in a memory;

We already say this for sure!!

(a-3) calculating said tap coefficient $\mathbf{G}(n)$ based on said received signal $\mathbf{R}(n)$ and said known signal, storing a tap coefficient $\mathbf{G}'(n)$ for said linear filtering corresponding to said effective path in a memory, and storing a received signal vector $\mathbf{R}'(n)$ corresponding to said effective path in a memory; and

(a-4) calculating said first soft decision value $\lambda_1[b(n)]$ from said estimated impulse response value $\mathbf{H}_m(n)$, said tap coefficient $\mathbf{G}'(n)$ and said received signal vector $\mathbf{R}'(n)$.

G(n) not G'(n)

6. The method of claim 5, wherein:

said step (c) comprises the steps of:

(c-1) calculating the tap coefficient $\mathbf{G}'(n)$ for said linear filtering corresponding to said effective path from said estimated impulse response value $\mathbf{H}_L'(n)$ composed of the component corresponding to said effective path through utilization of an Matrix Inversion Lemma; and

(c-2) storing said tap coefficient $\mathbf{G}'(n)$ for said linear filtering corresponding to said effective path in a memory;

said step (d) is a step of linear-filtering said tap coefficient $\mathbf{G}'(n)$ with

said estimated impulse response value vector $\mathbf{H}_L'(n)$ to obtain a replica signal;

said step (e) is a step of storing in a memory a difference signal $\mathbf{R}_c'(n)$ corresponding to said effective path, said difference signal $\mathbf{R}_c'(n)$ being obtained by subtracting said replica signal from said received signal $\mathbf{R}'(n)$;

said step (f) is a step of linear-filtering said difference signal $\mathbf{R}_c'(n)$ with said rap coefficient $\mathbf{G}'(n)$ to generate a signal $\mathbf{Z}'(n)$; and

said step (g) is a step of calculating said first soft decision value $\lambda_1[b(n)]$ from said estimated impulse response value vector $\mathbf{H}_L'(n)$ and said signal $\mathbf{Z}'(n)$.

7. The method of claim 2, wherein, letting J represent the maximum number of delayed symbols to be considered, and letting a received signal sample vector of an M-channel channel be represented by $\mathbf{r}(n)=[r_0(n)r_1(n)\ldots r_{M-1}(n)]^T$, said received signal vector $\mathbf{R}(n)$ by $\mathbf{R}(n)=[\mathbf{r}(n+J-1)\mathbf{r}(n+J-2)\ldots\mathbf{r}(n)]^T$, a channel weighting coefficient vector by $\mathbf{h}(n;j)=[h_0(n;j)h_1(n;j)\ldots h_{M-1}(n;j)]^T$, and a channel matrix $\mathbf{H}_m(n)$ of said estimated impulse response value by

$$H_m(n)=\begin{bmatrix} \mathbf{h}(n;0) & \mathbf{h}(n;1) & \cdots & \mathbf{h}(n;J-1) & \mathbf{0} & \cdots & \mathbf{0} \\ \mathbf{0} & \mathbf{h}(n;0) & \mathbf{h}(n;1) & \cdots & \mathbf{h}(n;J-1) & \mathbf{0} & \mathbf{0} \\ \vdots & & & \ddots & & & \vdots \\ \mathbf{0} & \mathbf{0} & \mathbf{h}(n;0) & \mathbf{h}(n;1) & \cdots & \mathbf{h}(n;J-1) & \mathbf{0} \\ \mathbf{0} & \cdots & \mathbf{0} & \mathbf{h}(n;0) & \mathbf{h}(n;1) & \cdots & \mathbf{h}(n;J-1) \end{bmatrix}_{MJ\times(2J-1)}$$

said step (a) comprises the steps of:

(a-1) linear-filtering a training signal b(n) in a training signal period with said estimated impulse response value $\mathbf{H}_m(n)$ to generate a replica $\mathbf{H}_m(n)b(n)$;

(a-2) generating the difference between said received signal $\mathbf{R}(n)$ and said replica $\mathbf{H}_m(n)b(n)$ as a difference vector $\mathbf{R}_c(n)$;

(a-3) linear-filtering said received signal $\mathbf{R}(n)$ with said tap coefficient

$\mathbf{G}(n)$ to generate an output $\mathbf{Z}(n)=\mathbf{G}(n)^H\mathbf{R}(n)$;

(a-4) determining said tap coefficient $\mathbf{G}(n)$ by an adaptive algorithm based on the difference between said output $\mathbf{Z}(n)$ and said training signal b(n); and

(a-5) calculating a soft decision value $\lambda_1[b(n)]=4\text{Real}\{\mathbf{Z}(n)\}/(1-\mu)$ based on the estimated impulse response value $\mathbf{H}_m(n)$ for linear filtering and said output $\mathbf{Z}(n)$, and outputting said soft decision value $\lambda_1[b(n)]=4\text{Real}\{\mathbf{Z}(n)\}/(1-\mu)$ as the result of said first round of equalization;

said step (b) is a step of calculating the likelihood $b'(k)=\tanh[\lambda_2[b(k)/2]$ of a code bit sequence b(k) from a soft decision value $\lambda_2[b(n)]$ of a decoded bit provided as a priori information with said k set within the range of $n-(j-1)\leq k\leq n+(J-1)$;

said step (c) is a step of calculating said tap coefficient $\mathbf{G}(n)$ by approximating

$$G(n)=[H_L(n)H_L(n)^H(n)-\sigma^2 I]^{-1}H_L(n)$$

$$H_L(n)=[h_0(n;J-1)\ldots h_{M-1}(n;J-1)h_0(n;J-2)\ldots h_{M-1}(n;J-2)\ldots h_0(n;0)\ldots h_{M-1}(n;0)]^T$$

said step (d) is a step of linear-filtering, with said $\mathbf{H}_L(n)$, an estimated value vector,

$$\mathbf{B}'(n)=[b'(n+(J-1))b'(n+(J-2))\ldots b'(n+1)0b'(n-1)\ldots b'(n-(J-1))]^T,$$

of a code bit that affects, as intersymbol interference, said code bit b(n) at time n to thereby obtain a replica $\mathbf{H}_L(n)\mathbf{B}'(n)$;

said step (e) is a step of calculating a difference vector $\mathbf{R}_c(n)=\mathbf{R}(n)-\mathbf{H}_L(n)\mathbf{B}'(n)$ between said replica $\mathbf{H}_L(n)\mathbf{B}'(n)$ and said received signal $\mathbf{R}(n)$;

said step (f) is a step of linear-filtering said difference vector $\mathbf{R}_c(n)$ with said tap coefficient $\mathbf{G}(n)$ and outputting the result of said linear filtering $\mathbf{Z}(n)=\mathbf{G}(n)^H\mathbf{R}_c(n)$; and

said step (g) is a step of obtaining a soft decision value

$$\lambda_1[b(n)] = \frac{4\mathrm{Real}\{Z(n)\}}{1-\mu(n)}$$

$$\mu(n)=H_L(n)^H G(n)$$

as the output of said second and subsequent rounds of equalization from said output $\mathbf{Z}(n)$ and said estimated impulse response value vector $\mathbf{H}_L(n)$.

8. The method of claim 6, wherein, letting J represent the maximum number of delayed symbols to be considered, and letting a received signal sample vector of an M-channel channel be represented by $\mathbf{r}(n)=[r_0(n)r_1(n)\ldots r_{M-1}(n)]^T$, said received signal vector $\mathbf{R}(n)$ by $\mathbf{R}(n)=[\mathbf{r}(n+J-1)\mathbf{r}(n+J-2)\ldots\mathbf{r}(n)]^T$, a channel weighting coefficient vector by $\mathbf{h}(n;j)=[h_0(n;j)h_1(n;j)\ldots h_{M-1}(n;j)]^T$, and a channel matrix $\mathbf{H}_m(n)$ of said estimated impulse response value by

$$H_m(n) = \begin{bmatrix} \mathbf{h}(n;0) & \mathbf{h}(n;1) & \cdots & \mathbf{h}(n;J-1) & \mathbf{0} & \cdots & \mathbf{0} \\ \mathbf{0} & \mathbf{h}(n;0) & \mathbf{h}(n;1) & \cdots & \mathbf{h}(n;J-1) & \mathbf{0} & \mathbf{0} \\ \vdots & & & \ddots & & & \vdots \\ \mathbf{0} & \mathbf{0} & \mathbf{h}(n;0) & \mathbf{h}(n;1) & \cdots & \mathbf{h}(n;J-1) & \mathbf{0} \\ \mathbf{0} & \cdots & \mathbf{0} & \mathbf{h}(n;0) & \mathbf{h}(n;1) & \cdots & \mathbf{h}(n;J-1) \end{bmatrix}_{MJ\times(2J-1)}$$

said step (a-3) comprises the steps of:

(a-3-1) linear filtering a training signal b(n) in a training signal period with said estimated impulse response value $\mathbf{H}_m(n)$ to generate a replica $\mathbf{H}_m(n)b(n)$;

(a-3-2) generating the difference between said received signal $\mathbf{R}(n)$ and said replica $\mathbf{H}_m(n)b(n)$ as a difference vector $\mathbf{R}_c(n)$;

(a-3-3) linear filtering said received signal $\mathbf{R}(n)$ with said tap coefficient $\mathbf{G}(n)$ to generate an output $\mathbf{Z}(n)=\mathbf{G}(n)^H\mathbf{R}(n)$; and

(a-3-4) determining said tap coefficient $\mathbf{G}(n)$ by an adaptive algorithm based on the difference between said output $\mathbf{Z}(n)$ and said training signal b(n),

and storing a received signal $\mathbf{R}'(n)$ and a tap coefficient $\mathbf{G}'(n)$ of those components of said received signal $\mathbf{R}(n)$ and said tap coefficient $\mathbf{G}(n)$ which correspond to said effective path;

said step (a-4) is a step of calculating a soft decision value $\lambda_1[b(n)]=4\mathrm{Real}\{\mathbf{Z}(n)\}/(1-\mu)$ based on the estimated impulse response value $\mathbf{H}_m'(n)$ for said linear filtering and said output $\mathbf{Z}(n)$, and outputting said soft decision value $\lambda_1[b(n)]=4\mathrm{Real}\{\mathbf{Z}(n)\}/(1-\mu)$ as the result of said first round of equalization;

said step (b) is a step of calculating the likelihood $b'(k)=\tanh[\lambda_2[b(k)/2]$ of a code bit sequence $b(k)$ from a soft decision value $\lambda_2[b(n)]$ of a decoded bit provided as a priori information with said k set within the range of $n-(j-1)\leq k\leq n+(J-1)$;

said step (c-1) is a step of calculating said tap coefficient $\mathbf{G}'(n)$ by approximating

$$G'(n)=[H_L'(n)H_L'(n)^H(n)-\sigma^2I]^{-1}H_L'(n)$$

$$H_L'(n)=[h_0(n;J-1)\ldots h_{M-1}(n;J-1)h_0(n;J-2)\ldots h_{M-1}(n;J-2)\ldots h_0(n;0)\ldots h_{M-1}(n;0)]^T$$

said step (d) is a step of linear filtering, with said HL'(n), an estimated value vector,

$$\mathbf{B}'(n)=[b'(n+(J-1))b'(n+(J-2))\ldots b'(n+1)0b'(n-1)\ldots b'(n-(J-1))]^T,$$

of a code bit that affects, as intersymbol interference, said code bit $b(n)$ at time n to thereby obtain a replica $\mathbf{H}_L(n)\mathbf{B}'(n)$;

said step (e) is a step of calculating a difference vector $\mathbf{R}_{c'}(n)=\mathbf{R}'(n)-\mathbf{H}_{L'}(n)\mathbf{B}'(n)$ between said replica $\mathbf{H}_L'(n)\mathbf{B}'(n)$ and said received signal $\mathbf{R}'(n)$;

said step (f) is a step of linear-filtering said difference vector $\mathbf{R}_c'(n)$ with said tap coefficient $\mathbf{G}'(n)$ and outputting the result of said linear filtering $\mathbf{Z}'(n)=\mathbf{G}'(n)^H\mathbf{R}_c'(n)$; and

said step (g) is a step of obtaining a soft decision value

$$\lambda_1[b(n)] = \frac{4\mathrm{Real}\{Z'(n)\}}{1-\mu(n)}$$

$$\mu(n)=H_L'(n)^H G'(n)$$

as the output of said second and subsequent rounds of equalization from said output $\mathbf{Z}'(n)$ and said estimated impulse response value vector $\mathbf{H}_L'(n)$.

9. An adaptive equalizer comprising:

an impulse response estimating part for calculating an estimated impulse response value of each of Channel based on a received signal and a known signal;

a tap coefficient calculating part for calculating the tap coefficient of a linear filter by an adaptive algorithm based on said received signal and said known signal;

said linear filter having set therein said tap coefficient, for linear-filtering said received signal; and

a soft decision value calculating part for calculating a soft decision value from said estimated impulse response value and the result of said linear filtering.

10. The equalizer of claim 9 further comprising:

a storage part for storing said estimated impulse response value;

a likelihood calculating part for calculating, from said soft decision value, its likelihood; and

means for obtaining a replica signal by subjecting said likelihood to linear filtering with an estimated impulse response value vector obtained from said stored estimated impulse response value vector by approximating intersymbol interference with the code b(n) concerned to zero; ?! prior art prior art

wherein said tap coefficient calculating part includes means for

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calculating the tap coefficient of said linear filter by an adaptive algorithm based on said received signal and said known signal when a soft decision value is input from a decoder to said tap coefficient calculating part, and for calculating said tap coefficient from said estimated impulse response value vector when said soft decision value is input to said tap coefficient calculating part; and

said linear filter performs linear filtering of said received signal in the absence of said soft decision value and the difference between said received signal and said replica signal through the use of said tap coefficient in the presence of said soft decision value, and provides the linear filtering output to said soft decision value calculating part.

11. The adaptive equalizer of claim 9, further comprising:

a path decision part for determining from said estimated impulse response value of said each channel whether the received power of the corresponding path is larger than a predetermined reference value;;

a path memory for storing as an effective path a path determined as being larger than said predetermined reference value;

a tap coefficient memory for storing as a new tap coefficient the component of that one of said tap coefficients corresponding to said effective path; and

a signal identifying part for identifying that received signal component of said received signal corresponding to said effective path;

wherein said soft decision value calculating part calculates said soft decision value from the received signal component corresponding to said effective path, the estimated impulse response value corresponding to said effective path and the tap coefficient corresponding to said effective path.

12. The adaptive equalizer of claim 11, further comprising:

a likelihood calculating part for calculating the likelihood of a code from said soft decision value;

a replica generating linear filter for generating a replica signal of said received signal by linear-filtering said likelihood with an estimated impulse response value composed of that component of said estimated response value vector corresponding to said effective path;

a subtractor for subtracting said replica signal from that component of said received signal corresponding to said effective path to obtain a difference signal; and

a difference memory for storing only that difference signal component of said difference signal corresponding to said effective path;

wherein sad tap coefficient calculating part calculates, in second and subsequent rounds of equalization, the tap coefficient of said linear filter from said estimated impulse response value vector corresponding to said effective path through the use of a Matrix Inversion Lemma; and

said soft decision calculating part is a means for calculating, in said second and subsequent rounds of equalization, said soft decision value from said estimated impulse response value vector corresponding to said effective path, said tap coefficient of said linear filter corresponding to said effective path and said difference signal corresponding to said effective path.

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